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(54) Adaptive equalization for recording systems using partial-response signaling.

(57) The channel characteristics of magnetic-disk storage devices vary with track radius. An adaptive 3-tap transversal equalizer that compensates these variations for systems using partial-response signaling is presented. The equalizer coefficients are updated by applying a procedure that is related to the recursive least-squares algorithm. This new updating procedure does not require multiplications and is well suited for high-speed implementation. Results obtained by computer simulations and measurements with a prototype have shown that the proposed adaptive equalizer can effectively compensate variations in magnetic-disk channel characteristics with track radius.

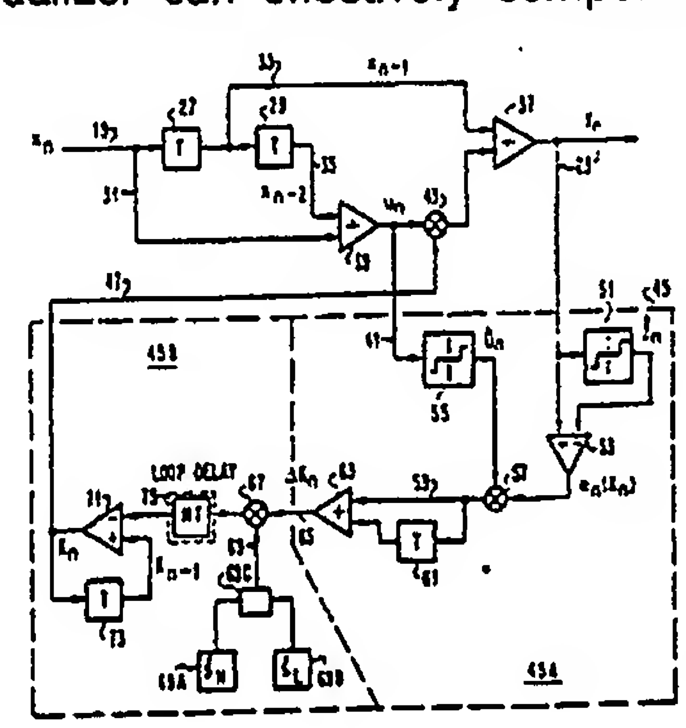

FIG. 4

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ADAPTIVE EQUALIZATION FOR RECORDING SYSTEMS USING PARTIAL-RESPONSE SIGNALING

FIELD OF INVENTION

Present invention relates to recording systems using partial-response signaling, e.g. magnetic disk storage systems or optical disk storage systems using partial-response signaling with maximum-likelihood sequence detection (PRML). The invention relates in particular to a method for updating the tap coefficients of an equalizer in such systems, allowing to compensate the effect of channel characteristic variations with track radius, and further to an equalizer for executing this method.

BACKGROUND

In disk storage devices using partial-response (PR) signaling, a receive filter in the readback apparatus has to shape the output signal of the recording channel into a PR signal before a detection device, e.g. a maximum-likelihood sequence detector can reconstruct the recorded data sequence. However, since the recording density of data on a track, and thus the recording channel characteristic varies with track radius, a fixed filter can only shape the channel output signal into a nominal PR signal at a given track; at other radii, the spectrum of the filter output signal deviates from the nominal PR characteristic leading to a degradation in error-rate performance of the recording system. Such a loss in performance can be avoided if an adaptive equalizer is used besides the fixed receive filter, to compensate for variations in channel characteristics with track radius.

Various equalization circuits and methods for improving the operability and effectiveness of recording or communication systems are known in the art.

In the publication "Improvement of recording density by means of a cosine equalizer" by T.Kameyama et al., IEEE Transactions on Magnetics, Vol.MAG-12, No.6 (Nov.1976), pp.746-748, an equalizer was disclosed which improves the recording density in a peak detection magnetic recording system by pulse slimming. The equalizer consists of a delay line, an amplitude divider and a differential amplifier. However, the equalizer is not adaptively updated during reading data from the disk and thus cannot compensate channel characteristic variations.

An article by D.D.Falconer et al. entitled "Application of fast Kalman estimation to adaptive equalization", published in the IEEE Transactions on Communications, Vol.COM-26, No.10 (Oct.1978), pp.1439-1446, suggested to employ, for the setting of tap coefficients in an adaptive equalizer in data communication systems, a recursive least squares algorithm. This algorithm leads to rapid initial convergence of the equalizer tap coefficients. However, the algorithm requires a great number of multiplications and thus is not suited for high-speed implementation.

U.S.Patent 4,580,176 entitled "Adaptive equalization circuit for magnetic recording channels utilizing signal timing", and the article "Adaptive symmetrical interference equalization" by R.C.Schneider et al., IBM Technical Disclosure Bulletin Vol.28, No.11 (April 1986), pp.4857-4858, disclose adaptive equalizer circuits for magnetic recording systems. However, they are designed for peak detection systems and thus are not suited for recording systems using partial-response signaling.

OBJECTS OF THE INVENTION

It is a primary object of the invention to compensate channel characteristic variations with track radius in recording systems using partial-response signaling.

It is a further object of this invention to devise a tap coefficient adjustment method for an equalizer of a recording system using partial-response signaling, by which the effect of channel characteristic variations with track radius is minimized.

Another object of the invention is to devise circuitry for the adaptive adjustment of the equalizer tap coefficients which is well suited for high-speed implementation.

A further object is a method of equalizer tap coefficient adjustment which enables a fast start-up of the equalizer circuitry in recording systems using partial-response signaling.

SUMMARY OF THE INVENTION

These objects are achieved by a method for tap coefficient adjustment in an equalizer of a recording system using partial-response signaling, as defined in Claim 1 and Claim 6, and by equalizer apparatus for executing this method as defined in Claim 7.

The invention compensates the effects of varying channel characteristics due to varying radius in recording systems using PR signaling, and allows fast initial adjustment (during training sequence reception) and reliable updating (during data reception) of the equalizer coefficients. Since only logic operations and no multiplications are necessary to update the tap coefficients, the equalizer is well suited for high-speed implementation.

Further features and advantages of the invention will become apparent from the following detailed description of a preferred embodiment in connection with the accompanying drawings.

LIST OF DRAWINGS

Fig.1 is a schematic overview of a PRML recording system with an adaptive cosine equalizer (ACE);

Fig.2 is a basic block diagram of the cosine equalizer;

Fig.3 is a diagram giving amplitude spectra of the cosine equalizer and of the ideal reference equalizer.

Fig.4 is a block diagram of the adaptive cosine equalizer with tap coefficient adjustment circuitry according to the invention;

Fig.5 is a block diagram of a digital implementation of the basic cosine equalizer; and

Fig.6 is a block diagram of a digital implementation of the tap coefficient gradient ($\Delta K_n/2$) computation.

DETAILED DESCRIPTION

1. PRML System with ACE

In the following, an embodiment of the invention is described for the example of a disk storage system having the following characteristics: It is a magnetic recording system using partial-response class-IV signaling with maximum-likelihood sequence detection (PRML).

A PRML system with an adaptive cosine equalizer (ACE) is shown in Figure 1. A binary data sequence $\{a_n = \pm 1\}$ is sent at the signaling rate 1/T through a magnetic-disk storage system 11, band-limiting receive filter 13, and variable gain amplifier (VGA) 15 with gain γ. The output of VGA 15 is sampled by sampling means 17, e.g. an analog-to-digital converter, at times $nT + \tau$ and the resulting samples x_n appearing on line 19 are processed by ACE 21. ACE 21 is a symmetric 3-tap digital equalizer with a center-tap value set to one. The other two taps are adjustable and have the same value K. The ACE will be described in more detail later. VGA 15 and ACE 21 together constitute a 3-tap transversal filter with coefficients γK, γ, and γK. The output samples y_n appearing on the output lines 23 of the ACE are processed by Viterbi decoder 25 to reconstruct the recorded data sequence.

For a PR-IV signaling scheme,

$y_n = z_n + w_n$, (1)

where

$z_n = a_n - a_{n-2}$ (2)

represents the signal part and w_n is filtered noise. The fixed receive filter can shape the output of the magnetic-disk channel into the form described by (1) and (2) only at a given track radius. By adjusting the gain γ and ACE coefficient K, this form can be closely approximated at other track radii.

The effectiveness of the ACE to compensate variations of the magnetic-disk channel characteristics with track radius will be shown using a model of these variations derived from measurements. Following this model, the overall transfer characteristic of the magnetic-disk channel and receive filter is given by

$C(f) = \eta\, C_0(f)\, e^{\Delta/2}\, e^{-2T\Delta|f|}$, (3)

where

$$C_0(f) = \begin{cases} T\,[1 - e^{-j4\pi fT}] & \text{for} \quad |f| < 1/2T \\ 0 & \text{otherwise} \end{cases}, \tag{4}$$

is the transfer characteristic of a PR-IV system. In (3), the gain parameter η. and the distortion parameter Δ model deviations from the PR-IV transfer characteristic caused by variations of the magnetic-disk channel characteristics with track radius. The term $e^{\Delta/2}$ in (3) normalizes the transfer characteristic C(f) so that at f = 1/4T, C(1/4T) = $\eta 2T$ is independent of the distortion parameter Δ.

2. Cosine Equalizer

A block diagram of a cosine equalizer is shown in Figure 2. It comprises a delay line having two delay elements 27, 29 and three taps 31, 33, 35. The sample of the center tap and modified samples of the two outer taps, each of them weighted with tap coefficient K, are combined in an adder 37 to form output sample y_n. The output sample y_n is given by

$y_n(K) = \tilde{x}_n + K\,u_n$, (5)

where

$\tilde{x}_n = x_{n-1}$ (6)

denotes the sample at the center of the delay line and

$u_n = \tilde{x}_{n+1} + \tilde{x}_{n-1}$. (7)

The filter described by (5) and (7) with $\tilde{x}_n$ as input has no phase shift; its transfer function is

$H(f) = 1 + 2K\cos(2\pi fT)$. (8)

The transfer function of a reference equalizer that exactly compensates variations in channel characteristics for the model described by (3) and (4) is given by

$H_0(f) = e^{-\Delta/2}\, e^{2T\Delta|f|}$. (9)

In Figure 3, the amplitude spectrum $|H(f)|$ of the cosine equalizer is compared to $|H_0(f)|$ of this reference equalizer for three distortion parameters Δ. The value of the VGA gain γ is chosen such that $\eta\gamma = 1$ and the coefficient K is selected to minimize the mean-squared error at the input of the Viterbi decoder. The discrepancy between $|H(f)|$ and $|H_0(f)|$ at DC and the Nyquist frequency does not result in a large mean-squared error since PR-IV signaling leads to spectral nulls at these frequencies.

3. Tap Adjustment Procedure

The tap adjustment scheme is derived from the fast recursive least squares (RLS) algorithm for the estimation of the optimal tap coefficient (as descibed in the paper by Falconer et al. mentioned above). At time n+1 the tap coefficient K_{n+1}^{RLS} obtained by the RLS method is the tap value K that minimizes the cumulative squared error

$$\sum_{m=0}^{n} e_m^2(K), \tag{10}$$

where

$e_m(K) = y_m(K) - \hat{z}_m$ (11)

represents the error signal at the output of the cosine equalizer with tap coefficient K and $\hat{z}_m$ denotes a reconstruction of z_m (see (2)) made by the receiver. The tap coefficient K_{n+1}^{RLS} can be generated recursively as follows (RLS algorithm):

$K_{n+1}^{RLS} = K_n^{RLS} - \xi_n e_n(K_n^{RLS})\,u_n$ (12)

$K_0 = 0$, (13)

where

$$\xi_n = \frac{1}{\sum_{m=0}^{n} u_m^2} \tag{14}$$

is a time varying loop gain.

The signal sample z_n (2) can only have the values 0, ±2 and (1) suggests to reconstruct the three-level sample

$$\hat{z}_n = \begin{cases} 2 & \text{for} \quad 1 \le y_n \\ 0 & \text{for} \quad -1 < y_n < 1 \\ -2 & \text{for} \quad y_n \le -1 \ . \end{cases} \tag{15}$$

The term $e_n\, u_n$ in (12) is recognized as the stochastic gradient $1/2\ de_n^2(K_n)\ dK_n$ of the mean squared error $E[e_n^2(K_n)]$ with respect to the tap coefficient K_n. Computation of this gradient requires a multiplication that can be avoided by using, according to the invention, instead of u_n the three-level reconstruction

$$\hat{u}_n = \begin{cases} 2 & \text{for} \quad 1 \le u_n \\ 0 & \text{for} \quad -1 < u_n < 1 \\ -2 & \text{for} \quad u_n \le -1 \ . \end{cases} \tag{16}$$

This approximation is justified by noticing that when the signal at the output of the receive filter has already the PR-IV format, leading to K = 0, then (see (1), (2), (5), and (7))

$$u_n = y_{n+1} + y_{n-1} = a_{n+1} - a_{n-3} + w_{n+1} + w_{n-1} = \begin{cases} 2 + \text{noise} \\ 0 + \text{noise} \\ -2 + \text{noise} \ . \end{cases} \tag{17}$$

An approximate tap gradient can now be computed without multiplication. It is advantageous to compute the sum of two consecutive gradients in order to reduce variance. This leads to

$\Delta K_n = e_n(K_n)\, \hat{u}_n + e_{n-1}(K_{n-1})\, \hat{u}_{n-1}$.   (18)

This gradient is used to update the tap coefficient by:

$K_{n+1} = K_n - \xi \Delta K_n$.   (19)

In the RLS algorithm (12), the loop gain ξ_n computed by (14) decreases at every iteration. A simpler method that approximates this behavior is used for the modified algorithm (19). The loop gain ξ is first set to a high value ξ_H for fast initial adjustment of the tap coefficient and then lowered to a value ξ_L to reduce fluctuations in the tap adjustment caused by noise. The modified algorithm can track variations in channel characteristics, which is not the case for the RLS algorithm given that $\xi_n \rightarrow 0$ as $n \rightarrow \infty$. The simpler method of switching the loop gain once instead of changing it according to (14) does not result in a loss of convergence speed.

Initial adjustment of the tap coefficient can be accelerated by transmitting a training sequence which consists of the repeated 18-bit long sequence:

... +1 +1 -1 +1 +1 -1 -1 -1 -1 +1 +1 +1 +1 -1 -1 +1 -1 -1   (20)

Transmitting this sequence leads to well decoupled adjustments of the VGA gain and tap coefficient. Correct timing phase adjustment is also maintained.

4. Structure of an Adaptive Cosine Equalizer Using the Invention

Fig.4 shows the block diagram of an adaptive cosine equalizer which consists of a basic cosine equalizer and tap coefficient adjustment means implementing the tap coefficient updating method represented by equations (18) and (19).

The ACE includes: a delay line, comprising two delay elements 27 and 29 and having two outer taps 31, 35 and a center tap 33, the first outer tap being connected to line 19 for receiving the input sample x_n; an adder 39 for combining the samples x_n at tap 31 and x_{n-2} at tap 35 to provide their sum u_n on line 41; multiplying means 43 for multiplying sample sum u_n by tap coefficient K_n; an adder 37 for adding the product $K_n u_n$ to the sample x_{n-1} appearing at the center tap of the delay line, for generating the output sample y_n on output line 23; and a tap coefficient adjustment means 45 having two inputs connected to lines 23 and 41 for receiving y_n and u_n, and an output furnishing the tap coefficient K_n on line 47.

A digital implementation of the basic cosine equalizer will be briefly described in the next section in connection with Fig.5.

The tap coefficient adjustment means 45 which is an important part of the invented adaptive equalizer comprises the following elements in its first portion 45A (which furnishes tap coefficient gradient ΔK_n):

- a quantizer 51 connected to output line 23 for receiving output sample y_n, and furnishing on its output a reconstructed sample $\hat{z}_n$ of the nominal PR-IV sample z_n which is either 2, 0, or -2 according to (15);
- subtracting means 53 connected to receive the output sample y_n from line 23 and the reconstructed sample $\hat{z}_n$ from the output of quantizer 51, respectively, and furnishing their difference as error value $e_n(K_n)$ on its output;
- a quantizer 55 connected to line 41 for receiving the sample sum u_n of the samples x_n and x_{n-2} and furnishing on its output the reconstructed sum $\hat{u}_n$ which is a reconstruction of the sum of the nominal PR-IV samples z_n and z_{n-2}, and which is either 2, 0, or -2 according to (16);
- combining means 57 connected to the output of subtracting means 53 and of quantizer 55 for receiving $e_n(K_n)$ and $\hat{u}_n$, and furnishing at its output 59 the product of both. However, instead of an explicit multiplication, only a selection from the three values $-2e_n(K_n)$, 0 , or $+2e_n(K_n)$ need to be made because the reconstructed sum $\hat{u}_n$ is ternary.
- a delay element 61 connected to the output of combining means 57 for delaying the respective output value by one sampling period T;
- an adder 63 connected to the outputs of combining means 57 and of delay element 61, for producing at its output 65 tap coefficient gradient ΔK_n in accordance with (18).

A digital implementation of the first portion 45A of tap coefficient adjustment means 45 will be briefly described in the next section in connection with Fig.6.

Tap coefficient adjustment means 45 further comprises (as shown in the second portion 45B in Fig.4):

- multiplication means 67 for multiplying the tap coefficient gradient ΔK_n furnished on line 65, by a loop gain ξ applied to input 69. This loop gain ξ assumes first the higher and then the lower of two preselected values ξ_H (69A) and ξ_L (69B), the switchover (69C) between these two values being performed at a predetermined time. Good values for the two loop gain factors and the switchover time can be determined either by simulation or by trial operation of the circuitry. It should be noted that no explicit multiplication is required when the loop gain ξ is a multiple power of two; then the loop gain can be adjusted by changing the weight of all bits of the gradient ΔK_n (shifting).
- subtracting means 71 for forming a new equalizer tap coefficient K_n from a previous tap coefficient K_{n-1} and the weighted gradient $\xi \Delta K_{n-M}$, furnishing the new tap coefficient on line 47;
- a delay element 73 connected to line 47, for delaying the equalizer tap coefficient K_n by one sampling period and for furnishing its output K_{n-1} to subtracting means 71.

A loop delay shown at 75 in Fig.4 has been introduced in the tap adjustment means 45 to represent the inherent signal processing delay caused by the time required to compute a gradient ΔK_n and update the equalizer tap coefficient. In Fig.4 a loop delay of M sampling periods T is shown. The delay elements are of course not concentrated at one location as shown in Fig.4 but distributed as single pipeline registers at proper locations in the digital realization of means 45. The number M depends on the signaling rate 1/T and on the technology used to realize the ACE.

5. Digital Implementation of the Adaptive Cosine Equalizer

In the following, a digital implementation of the cosine equalizer and the first portion (45A) of the tap coefficient adjustment means 45 will be described. A digital realization of the second portion (45B) shown in

Fig.4 is straightforward and will not be discussed.

Fig.5 shows the circuit implementation of the basic cosine equalizer. The input sample x_n is quantized with six bits and represented in two's complement (TC) form. The weight of its least significant bit (LSB) is 0.125. It should be noted that the amount of half an LSB has to be added to the sample x_n to compensate a displacement intentionally introduced by the analog-to-digital converter adjustment. In principle, this can be done at once by physically introducing a "seventh" bit with weight 2^{-4} which is always set to '1'. However, to simplify significantly the hardware design by taking advantage of this representation, the correction term is not added before some other operations are performed on the received bits. A 7-bit full adder is used to compute the sample $u_n = x_n + x_{n-2}$. At this stage the "seventh" bit with weight 2^{-4} of x_n and x_{n-2} is taken into account. The sample u_n is multiplied by the tap coefficient K_n. K_n is quantized with 4 bits and its absolute value is limited to 0.25. The product is quantized with 6 bits and sign extended before being added to the sample x_{n-1} represented in offset binary (OB) form. Element 81 converts the sample x_{n-1} appearing on line 33 from TC form to OB representation. The resulting sum is furnished in offset binary representation because a TC and OB number are added. In case of underflow or overflow the output saturates to its minimum or maximum value. The saturation circuit consists of one EXOR gate and a multiplexer. It is activated when the most significant bit (MSB) of the TC sample at the input of adder 37 does not equal the carry output c_0 of the adder; in this case the carry determines the saturation value. The equalizer output sample y_n is represented in the same form as the received sample x_n.

Circuitry for a digital implementation of the first portion 45A of tap coefficient adjustment means 45 (cf. Fig.4) is shown in Fig.6. Both input samples y_n and u_n are represented in two's complement form and are provided by the cosine equalizer (cf. Fig.5) on lines 23 and 41. Relations (11), (15) and (16) show that the term $e_n\hat{u}_n/2$ can be computed as follows:

$$\frac{e_n\hat{u}_n}{2} = \begin{cases} y_n - 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \geq 1 \\ y_n & \text{for} \quad u_n \geq 1 \quad \text{and} \quad |y_n| < 1 \\ y_n + 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \leq -1 \\ 0 & \text{for} \quad |u_n| < 1 \\ -y_n + 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \geq 1 \\ -y_n & \text{for} \quad u_n \leq -1 \quad \text{and} \quad |y_n| < 1 \\ -y_n - 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \leq -1 \,. \end{cases} \qquad (21)$$

Since its absolute value is less than or equal to 1.9375, it can be represented by five bits. Its computation using a simple EXOR/AND circuit combination is shown in portion 83 of Fig.6. This portion includes a group of five EXOR gates 83A, one additional EXOR gate 83B, and a group of five AND gates 83C. Its output signal is determined by a binary control signal B1 indicating when $u_n < 0$, by a binary control signal B2 indicating when $|u_n| \geq 1$, and by a binary control signal B3 indicating when $|y_n| \geq 1$. These binary control signals appear on lines 85, 87, and 89, respectively. Their derivation from the incoming samples is shown in portions 91 and 93 of Fig.6. Depending on the sign of u_n (represented by the signal B1 on line 85), the five EXOR gates 83A invert or keep unchanged the five least significant bits of the sample y_n. If $|y_n| \geq 1$ (as indicated by the signal B3 on line 89), either the value +2 or -2 is added to y_n by inverting the bit on line 83D with the aid of EXOR gate 83B. If $|u_n| < 1$ (as indicated by the signal B2 on line 87), the output is forced to be zero by overwriting the EXOR output signals with zeros using the group of AND gates 83C. Two succeding terms $e_n\hat{u}_n/2$ (one delayed by a delay element 95) are added in a 6-bit adder 97. Two additional gates and a delay element, as shown at 99 in Fig.6 below the full adder 97, consider the "seventh" bit with weight 0.0625 of the two successive samples to be added. The resulting sum $\Delta K_n/2$ on line 65′ is fed to the tap coefficient accumulator as shown in Fig.4.

Claims

1. A method for tap coefficient adjustment in an equalizer of a recording system using partial-response signaling, said equalizer comprising a delay line with two outer taps and a center tap, the outer tap samples being multiplied by tap coefficient K_n, and an equalizer output sample y_n being generated by combining the

multiplied outer tap samples and the center tap sample, said method comprising the following steps:
- deriving a reconstructed sample $\hat{z}_n$ from said equalizer output sample y_n by a quantizing operation, the resulting reconstructed sample having any one of p different nominal values (e.g. +2, 0, -2);
- deriving a reconstructed sum $\hat{u}_n$ by adding the outer tap samples and quantizing the resulting sample sum $\hat{u}_n$ so that the reconstructed sum $\hat{u}_n$ has any one of q different predetermined values (e.g. +2, 0, -2); and
- combining said reconstructed sum $\hat{u}_n$, said reconstructed sample $\hat{z}_n$, and said output sample y_n, to generate a tap coefficient gradient ΔK_n for updating said tap coefficient K_n.

2. Tap coefficient adjustment method in accordance with Claim 1, said combining step further inluding the steps of:
- generating an error value e_n by subtracting said reconstructed sample $\hat{z}_n$ from said output sample y_n;
- multiplying said error value e_n and said reconstructed sum $\hat{u}_n$ by combinational logic operations; and
- delaying the resulting product, and adding a delayed and current resulting product.

3.Tap coefficient adjustment method in accordance with Claim 1 or 2, including the further step of:
- generating a training sequence of the form
+1 +1 -1 +1 +1 -1 -1 -1 -1 +1 +1 +1 +1 -1 -1 +1 -1 -1 ,
and
- furnishing said training sequence repeatedly to the input of said equalizer prior to furnishing a data signal to it, to attain fast initial tap coefficient adjustment.

4. Tap coefficient adjustment method in accordance with Claim 1 or 2 or 3, comprising the following additional steps:
- multiplying said tap coefficient gradient ΔK_n by a loop gain factor ξ, said loop gain factor being selected from two given gain values (ξ_H ,ξ_L), and
- initially selecting the high value (ξ_H) of said two gain values for fast initial adjustment, and
- subsequently selecting the low value (ξ_L) of said two gain values to reduce fluctuations in the tap adjustment caused by noise.

5. Tap coefficient adjustment method in accordance with Claim 4, wherein said two given gain values (ξ_H , ξ_L) are powers of 2, and wherein the multiplication of said tap coefficient gradient ΔK_n by said loop gain factor ξ is effected by a shifting operation.

6. A method for tap coefficient adjustment in an equalizer of a recording system using partial-response signaling, said equalizer comprising a delay line with two outer taps and a center tap, the samples appearing at the two outer taps being multiplied by a tap coefficient K_n, and an equalizer output sample y_n being generated by combining the multiplied outer tap samples and the center tap sample, said method comprising the following steps:
- generating a sample sum u_n by adding the samples x_n and x_{n-2} appearing on the two outer taps;
- deriving, from said sample sum u_n, a first binary indicator signal B1 indicating whether $u_n < 0$, and a second binary indicator signal B2 indicating whether $|u_n| \geq 1$;
- deriving, from said output sample y_n, a third binary indicator signal B3 indicating whether $|y_n| \geq 1$;
- combining by combinational logic operations including AND operations and EXOR operations, said output sample y_n and said first, second, and third binary indicator signals to generate a quantity A_n according to the rule

$$A_n = \begin{cases} y_n - 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \geq 1 \\ y_n & \text{for} \quad u_n \geq 1 \quad \text{and} \quad |y_n| < 1 \\ y_n + 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \leq -1 \\ 0 & \text{for} \quad |u_n| < 1 \\ -y_n + 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \geq 1 \\ -y_n & \text{for} \quad u_n \leq -1 \quad \text{and} \quad |y_n| < 1 \\ -y_n - 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \leq -1 \; ; \end{cases}$$

and
- adding said quantity A_n and a previously generated quantity A_{n-1} delayed by one sampling time, to obtain a tap coefficient gradient $\Delta K_n/2$.

7. Equalizer apparatus for executing the method of Claim 1 or 6, including:
- delay line means (27, 29) comprising two delay elements and having a first outer tap (31) connected to the input (19) of the equalizer, a center tap (33), and a second outer tap (35);
- first adding means (39) connected to the first and second outer taps, providing on its output (41) a sample sum u_n;
- multiplying means (43) connected to the output of said first adding means and to a tap coefficient line (47), providing on its output an intermediate value;
- second adding means (37) connected to said first outer tap and to the output of said multiplying means, its output being connected to the equalizer output line (23) for furnishing an equalizer output sample y_n; and
- tap coefficient adjustment means (45) having two inputs connected to the outputs (41, 23) of said first and second adding means, and having an output connected to said tap coefficient line (47).

8. Equalizer apparatus in accordance with Claim 7, in which said tap coefficient adjustment means (45) includes:
- tap coefficient gradient generation means (45A) for receiving said sample sum u_n and said equalizer output sample y_n from the outputs (41, 23) of said first and second adding means, respectively, furnishing on its output (65) a tap coefficient gradient ΔK_n; and
- tap coefficient updating means (45B) being connected to said tap coefficient gradient generation means for receiving said tap coefficient gradient ΔK_n, and furnishing a tap coefficient K_n to said tap coefficient line (47).

9. Equalizer apparatus in accordance with Claim 8, wherein said tap coefficient gradient generation means (45A) comprises
- combinational logic circuitry (83, 91, 93) for generating a quantity A_n in response to said sample sum u_n and said equalizer output sample y_n according to the rule

$$A_n = \begin{cases} y_n - 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \geq 1 \\ y_n & \text{for} \quad u_n \geq 1 \quad \text{and} \quad |y_n| < 1 \\ y_n + 2 & \text{for} \quad u_n \geq 1 \quad \text{and} \quad y_n \leq -1 \\ 0 & \text{for} \quad |u_n| < 1 \\ -y_n + 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \geq 1 \\ -y_n & \text{for} \quad u_n \leq -1 \quad \text{and} \quad |y_n| < 1 \\ -y_n - 2 & \text{for} \quad u_n \leq -1 \quad \text{and} \quad y_n \leq -1 \; ; \end{cases}$$

and
- delay means (95) and adding means (97) connected to the output of said combinational logic circuitry, for generating a tap coefficient gradient according to the rule $\Delta K_n/2 = A_n + A_{n-1}$.

10. Equalizer apparatus in accordance with Claim 8 or 9, wherein said tap coefficient updating means (45B) comprises means (67, 69, 69A, 69B) for executing shift operations on said tap coefficient gradient ΔK_n, resulting in a multiplication of said tap coefficient gradient initially by the first value, and subsequently after a predetermined number of sampling intervals T by the other value of two predetermined loop gain values, each of said two loop gain values being a multiple power of 2.

FIG. 1

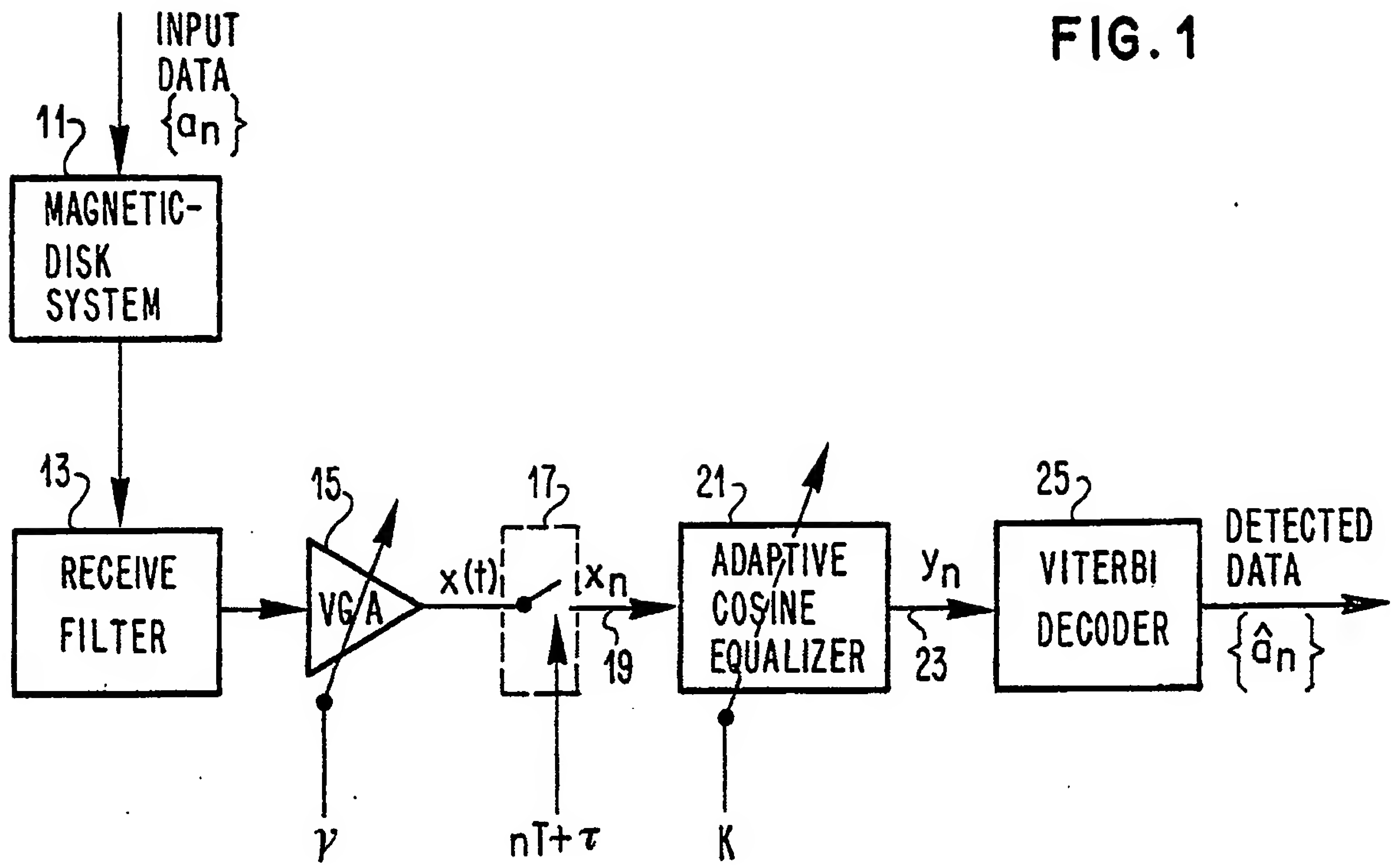
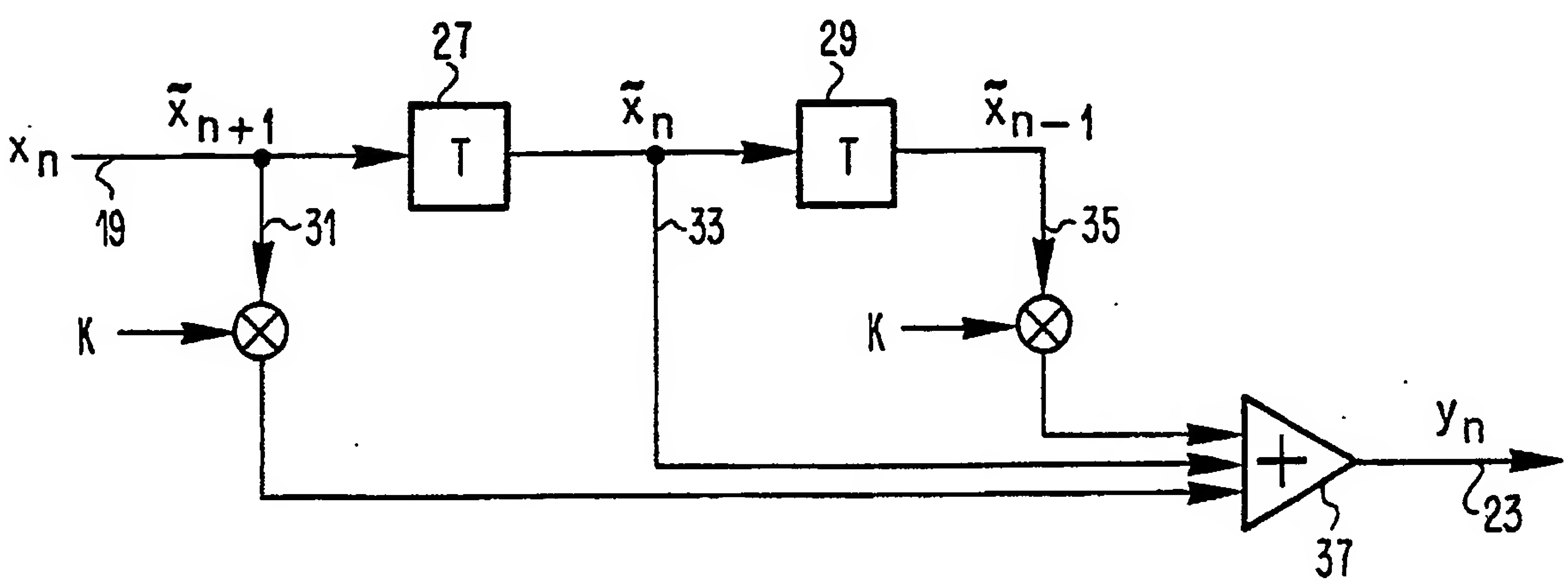
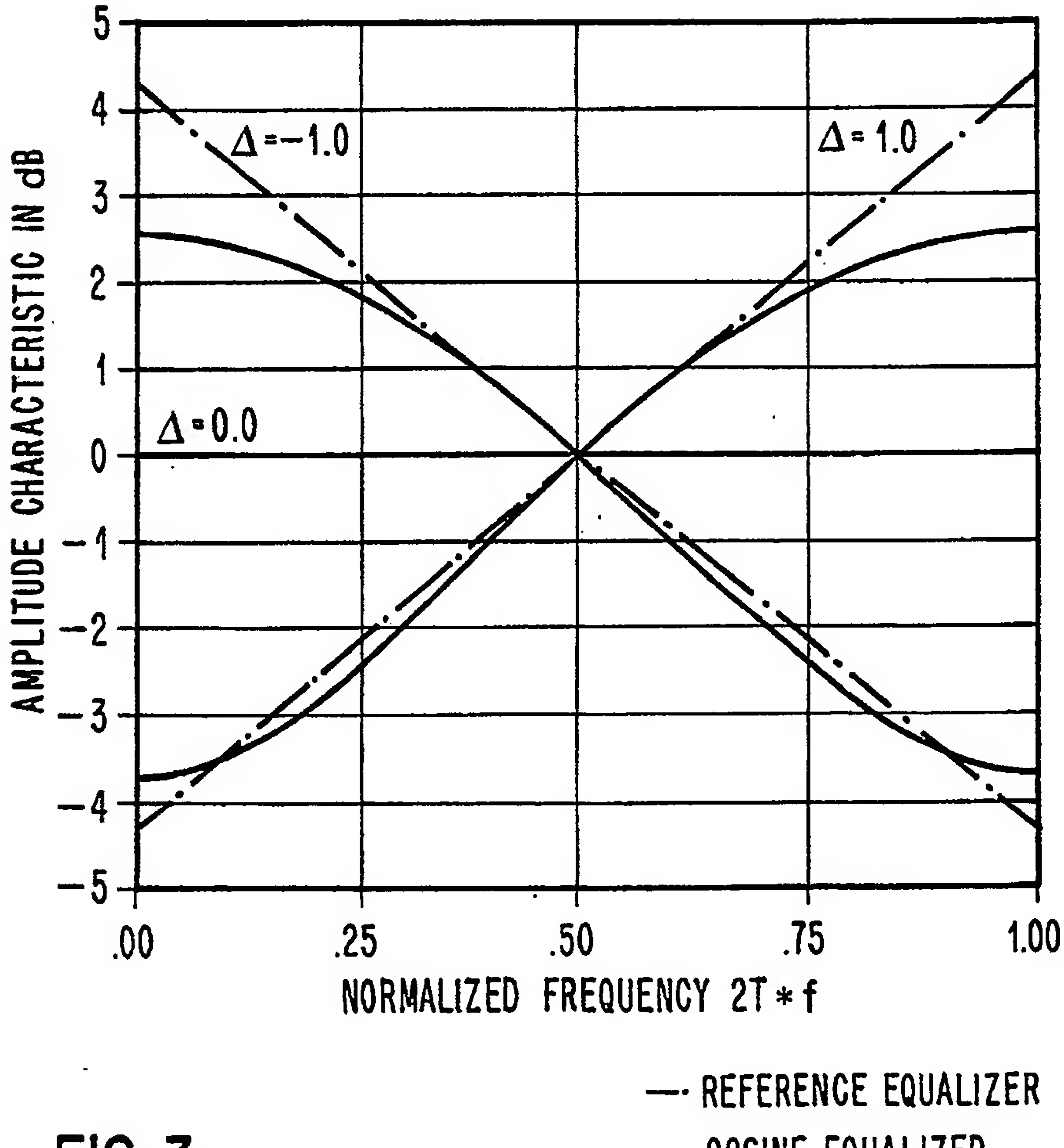

INPUT
DATA
{a_n}
11
MAGNETIC-
DISK
SYSTEM
13
RECEIVE
FILTER
15
VGA
x(t)
17
x_n
19
21
ADAPTIVE
COSINE
EQUALIZER
y_n
23
25
VITERBI
DECODER
DETECTED
DATA
{$\hat{a}_n$}
γ
$nT+\tau$
K

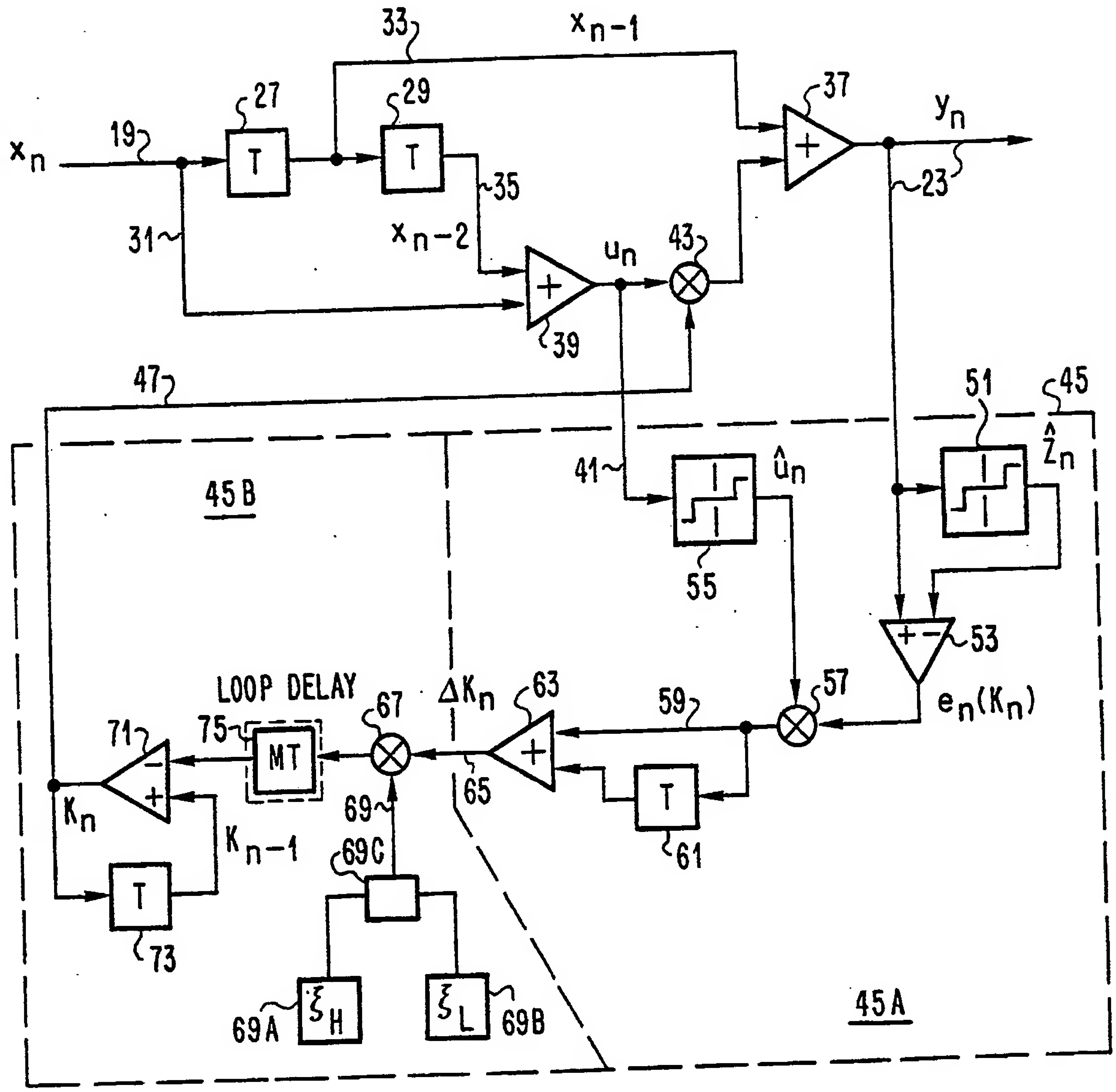
FIG. 2


x_n
19
$\tilde{x}_{n+1}$
31
27
T
$\tilde{x}_n$
33
29
T
$\tilde{x}_{n-1}$
35
K
K
+
37
y_n
23



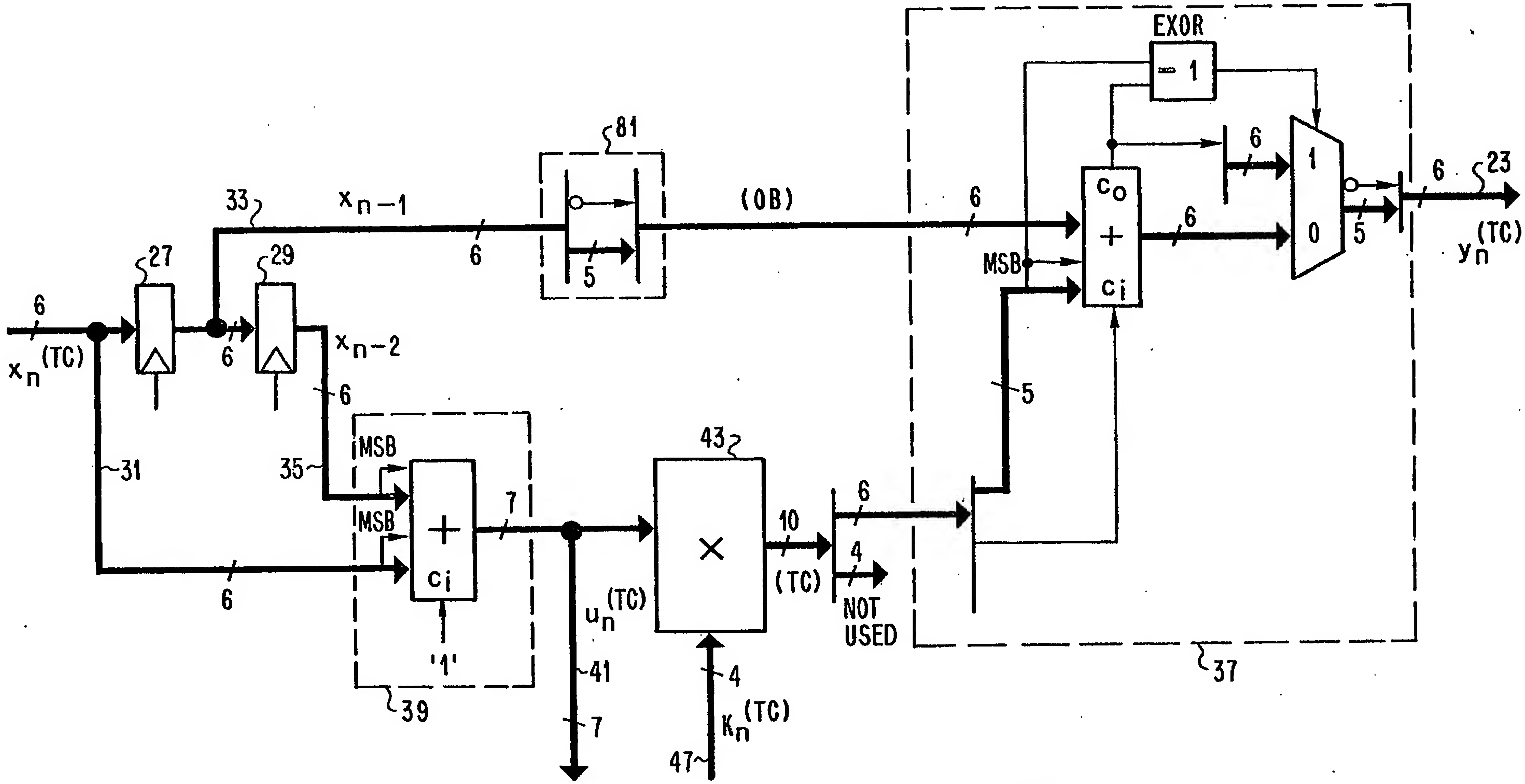
5
4
3
2
1
0
−1
−2
−3
−4
−5
AMPLITUDE CHARACTERISTIC IN dB
Δ=−1.0
Δ=1.0
Δ=0.0
.00
.25
.50
.75
1.00
NORMALIZED FREQUENCY 2T*f
—· REFERENCE EQUALIZER
— COSINE EQUALIZER

FIG. 3



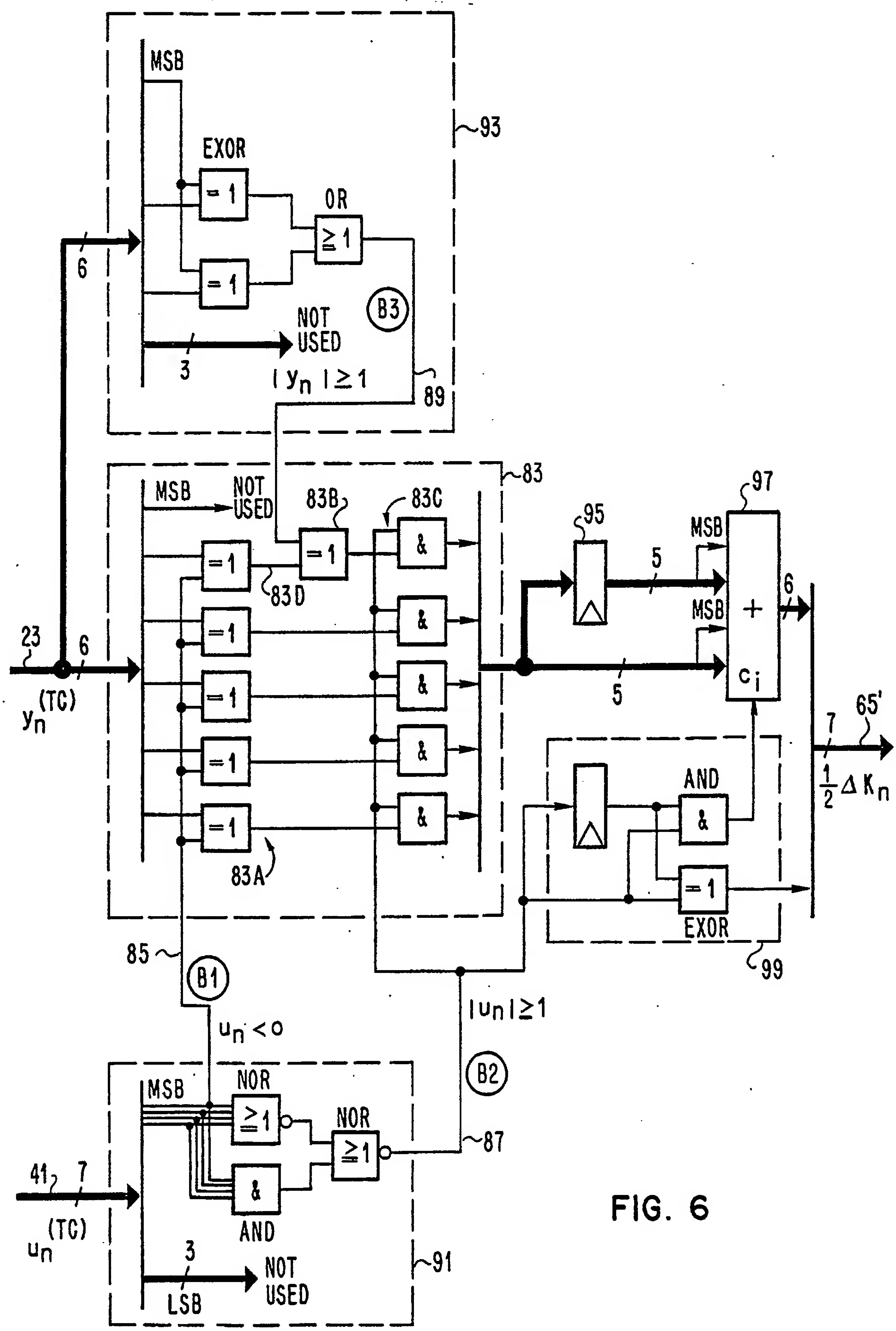
x_n
19
27
T
29
T
33
x_{n-1}
37
+
y_n
23
35
31
x_{n-2}
+
39
u_n
43
47
45
51
$\hat{Z}_n$
41
$\hat{u}_n$
55
45 B
53
$e_n(K_n)$
57
59
61
T
63
+
ΔK_n
65
67
LOOP DELAY
75
MT
71
K_n
K_{n-1}
T
73
69
69C
69A
ξ_H
ξ_L
69B
45 A

FIG. 4



6
$x_n^{(TC)}$
27
29
33
x_{n-1}
6
81
5
(OB)
x_{n-2}
6
35
31
6
MSB
MSB
7
c_i
'1'
39
43
10
(TC)
4
NOT USED
$u_n^{(TC)}$
41
7
4
$K_n^{(TC)}$
47
6
5
MSB
c_o
c_i
EXOR
= 1
6
6
1
0
5
6
23
$y_n^{(TC)}$
37

FIG. 5



MSB
EXOR
=1
OR
≥1
=1
93
6
NOT USED
B3
3
|yn|≥1
89
83
MSB
NOT USED
83B
83C
=1
&
83D
95
97
MSB
5
6
+
23
6
5
ci
yn(TC)
83A
7
65'
AND
1/2 ΔKn
EXOR
85
B1
99
un<o
|un|≥1
B2
MSB
NOR
NOR
87
41
7
AND
un(TC)
3
LSB
NOT USED
91

FIG. 6

European Patent Office

EUROPEAN SEARCH REPORT

Application Number

EP 88 81 0705

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int. Cl.5)
A	US-A-4564869 (H.P.BAUMEISTER) * column 2, line 51 - column 3, line 54; figures 1-3 * ---	1-2, 4, 7-8	G11B20/10 G11B5/035 H03H21/00
A	US-A-4041418 (H.KOETH) * column 2, line 43 - column 5, line 37; figures 2-3 * ---	1-2, 4, 7-8	
A	IEEE TRANSACTIONS ON COMMUNICATIONS. vol. COM34, no. 12, December 1986, NEW YORK US page 1272 - 1275; C.P.KWONG: "DUAL SIGN ALGORITHM FOR ADAPTIVE FILTERING" * page 1272, right-hand column, line 29 - page 1273, right-hand column, line 12; figures 1-2 * ---	1-2	
A	PATENT ABSTRACTS OF JAPAN vol. 10, no. 338 (P-516)(2394) 15 November 1986, & JP-A-61 139980 (FUJITSU LTD.) 27 June 1986, * see the whole document * ---		
A	DE-A-2013555 (SIEMENS AG.) ---		TECHNICAL FIELDS SEARCHED (Int. Cl.5)
A	US-A-3829780 (S.A.WHITE) ---		G11B H03H H03G H04L
A	IEEE TRANSACTIONS ON MAGNETICS. vol. MAG23, no. 5, September 1987, NEW YORK US page 3672 - 3674; SEIICHI MITA ET AL: "AUTOMATIC EQUALIZER FOR DIGITAL MAGNETIC RECORDING SYSTEMS" ---		
A	US-A-3676804 (K.H.MUELLER) -----		

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	21 JUNE 1989	CARTRYSSE A.A

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CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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